User's Manual

α Verb

24×32 Bit Digital Effects Module

www.altoproaudio.com
Version 2.0 Dec. 2002
— English —

SAFETY RELATED SYMBOLS

This symbol, wherever used, alerts you to the presence of un-insulated and dangerous voltages within the product enclosure. These are voltages that may be sufficient to constitute the risk of electric shock or death.

This symbol, wherever used, alerts you to important operating and maintenance instructions. Please read.

⏚ Protective Ground Terminal

~ AC mains (Alternating Current)

Hazardous Live Terminal

ON: Denotes the product is turned on.

OFF: Denotes the product is turned off.

WARNING

Describes precautions that should be observed to prevent the possibility of death or injury to the user.

CAUTION

Describes precautions that should be observed to prevent damage to the product.

WARNING

• Power Supply

Ensure that the mains source voltage (AC outlet) matches the voltage rating of the product. Failure to do so could result in damage to the product and possibly the user.

Unplug the product before electrical storms occur and when unused for long periods of time to reduce the risk of electric shock or fire.

• External Connection

Always use proper ready-made insulated mains cabling (power cord). Failure to do so could result in shock/death or fire. If in doubt, seek advice, from a registered electrician.

• Do not Remove any Covers

Within the product are areas where high voltages may present. To reduce the risk of electric shock do not remove any covers unless the AC mains power cord is removed.

Covers should be removed by qualified service personnel only.

No user serviceable parts inside.

• Fuse

To prevent fire and damage to the product, use only the recommended fuse type as indicated in this manual. Do not short-circuit the fuse holder. Before replacing the fuse, make sure that the product is OFF and disconnected from the AC outlet.

• Protective Ground

Before turning the product ON, make sure that it is connected to Ground. This is to prevent the risk of electric shock.
Never cut internal or external Ground wires. Likewise, never remove Ground wiring from the Protective Ground Terminal.

• Operating Conditions

Always install in accordance with the manufacturer's instructions.

To avoid the risk of electric shock and damage, do not subject this product to any liquid/rain or moisture. Do not use this product when in close proximity to water.

Do not install this product near any direct heat source.

Do not block areas of ventilation. Failure to do so could result in fire.

Keep product away from naked flames.

IMPORTANT SAFETY INSTRUCTIONS

Read these instructions

Follow all instructions

Keep these instructions. Do not discard.

Heed all warnings.

Only use attachments/accessories specified by the manufacturer.

- **Power Cord and Plug**

Do not tamper with the power cord or plug. These are designed for your safety.

Do not remove Ground connections!

If the plug does not fit your AC outlet seek advice from a qualified electrician.

Protect the power cord and plug from any physical stress to avoid risk of electric shock.

Do not place heavy objects on the power cord. This could cause electric shock or fire.

- **Cleaning**

When required, either blow off dust from the product or use a dry cloth.

Do not use any solvents such as Benzol or Alcohol.

For safety, keep product clean and free from dust.

- **Servicing**

Refer all servicing to qualified service personnel only.

Do not perform any servicing other than those instructions contained within the User's Manual.

Preface

Dear Customer,

Thanks for choosing ▲LTO α Verb and thanks for choosing one of the results of ▲LTO AUDIO TEAM job and researches.

For our ▲LTO AUDIO TEAM, music and sound are more than a job...are first of all passion and let us say...our obsession!

We have been designing professional audio products for a long time in cooperation with some of the major brands in the world in the audio field.

The ▲LTO line presents unparalleled analogue and digital products made by Musicians for Musicians in our R&D Centers in Italy, Netherlands, United Kingdom and Taiwan.

The core of our digital audio products is a sophisticated DSP (Digital sound processor) and a large range of state of the art algorithms which have been developed by our Software team for the last 7 years.

Because we are convinced you are the most important member of ▲LTO AUDIO TEAM and the one confirming the quality of our job, we'd like to share with you our work and our dreams, paying attention to your suggestions and your comments.

Following this idea we create our products and we will create the new ones! From our side, we guarantee you and we will guarantee you also in future the best quality, the best fruits of our continuous researches and the best prices.

Our ▲LTO α Verb is the result of many hours of listening and tests involving common people, area experts, musicians and technicians.

The result of this effort is the realization of effects such as reverb, chorus, flanger and delay that are today available in the best guitar amplifiers and studio equipment effects that we collected and transformed in presets now available in our small (half rack space), efficient and easy to use ▲LTO α Verb.

Nothing else to add, but that we would like to thank all the people that made the ▲LTO α Verb a reality available to our customers, and thank our designers and all the ▲LTO staff, there to make possible the realization of products containing our idea of music and sound and there to support you, our customers, in the best way, conscious that you are our best richness.

Thank you very much.
▲LTO AUDIO TEAM

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1. Introduction

Purchasing ▲LTO α Verb, you purchased a very powerful effect processor, easy to use and contained in a very efficient half rack package.

▲LTO α Verb is divided in 16 effects algorithms and 16 variations for each of these algorithms. The variations modify the most important parameter of the current algorithm and, for some algorithm as chorus and flanger, may decide also the shape of the modulating signal (Sinusoid or Ramp).

The first 8 algorithms are reverb algorithms and have been designed following the theory of different reverberation algorithms, taking care about the density and the body of the reverberated sound. The second 8 algorithms have been designed following the theory of modulation effects as chorus, flanger and rotary speakers. Within this second group of algorithms are included also the delays and the combined effects as flanger/ reverb, chorus/reverb and delay /reverb.

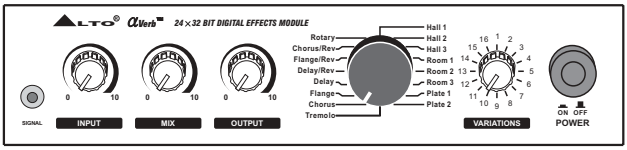
All the algorithms are based on classical algorithms for the effects generation and environment response modelling, modified and optimized thanks to the experience of ▲LTO AUDIO TEAM researchers.

2. Feature List

- Robust and Compact Design
- 24/32 bits Digital Audio Processor
- MPU Control
- Automatic Bypass Switch Detection
- 16 Great Sounding Programs
- Variation Adjust Knob (16 positions)
- Analog Mix (Dry/Wet) Potentiometer
- Variable Input Output Gain
- Stereo/Mono Jack Inputs
- Illuminated Power Switch
- Digital Saturation LED
- Up to 9dBu Line Level
- Easy to Operate Front Panel Controls
- SMT Design for Greater Reliability
- Short Signal Path and no Internal Cabling to Provide Superior Sound
- Manufactured Under QS9000, VDA6.1 Quality System

3. Front and Back Panels Description

3.1 Control Panel (Front Panel)



a. Program and Variations Selections

- **Program Select Knob:** The Program Select Knob is used to choose the program you wish to perform.
- **Variations Select Knob:** Each program on this apparatus has one parameter which can be adjusted. Depending on the type of program selected, this knob might alter reverb decay, chorus depth, etc.

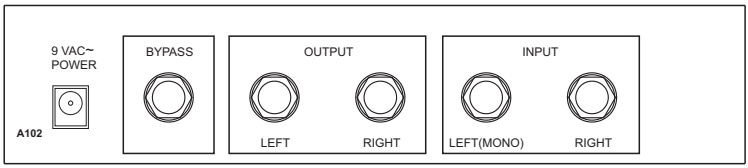
b. Analog levels

- **Analog Input Level Potentiometer:** The input level control sets the main input Gain, before the signal reaches the input bus. It controls both the Left (Mono) and Right input levels simultaneously.
- **Analog Output Level Potentiometer:** The output level control sets the level of the output signal going to the amplifier or mixer from α Verb.
- **Dry/Wet Mix Potentiometer:** Adjusts the balance between the dry signal coming into the input and the effects generated by α Verb.

c. Led and Illuminated Power Switch

- **Digital Saturation LED:** Displays the signal level coming into the input during Normal operation, if the signal level is too high, this LED light and you will begin to hear the signal distortion.
- **Power On/Off Switch:** Turns the apparatus on and off.

3.2 Analog Connections (Back Panel)



a. Analog Inputs/Outputs

- **Inputs:** These are 1/4" unbalanced phone jacks which connect to sources such as the effects sends of mixing console. They may be used with nominal input level up to 9dBu.
  For mono application, use the Left / Mono input. The Left / Mono input jack is normal to the Right jack. This means that when nothing is plugged into the Right input jack, the signal present at the Left / Mono input is routed to the Right as well.
- **Outputs:** These are 1/4" unbalanced phone jacks which connect to devices such as the effects returns on a mixing console or power amplifier inputs.

b. Effects Bypass Pedal Input

- **Effects Bypass:** This is a 1/4" phone jack which connects to a footswitch (with latching), either normally-open or normally-closed. When the footswitch is in the state of "OFF", the function will be "Effects"; On the other hand, when the footswitch is in the state of "ON", the effect will be disabled.

c. Power Connector

- Power connector: This is a plug for connecting the 9VAC power supply adaptor provided by the manufacturer.

4. Installation & Connection

4.1 Audio Connections and Power Up

a. Audio Connections

The connections between the α Verb and the other audio devices have to be made using high quality cables so to prevent bad performances of the α Verb itself. So it should be good to use low-capacitance shielded cables with a flexible internal conductor. Connect the cables to the α Verb properly by observing the following precautions:

- Do not bundle audio cables with AC power cords.
- Do not place audio cables and α Verb, near sources of electromagnetic interference such as transformers, monitors, computers, etc.
- Always unplug cables by firmly grasping the body of the plug and pulling directly outward.
- Do not place cables where they can be stepped on.
- Avoid twisting the cable or having it make sharp, right angle turns.

b. Power Up Setting

After making your connections, turn on the system's power using this procedure:
Before turning on the α Verb's power, check if:

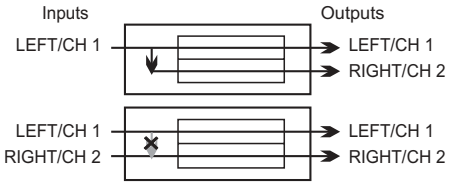
- All connections have been made correctly.
- The volume controls of the amplifier or mixer are turned down.

Insert the Power plug into the POWER input on the rear panel of the α Verb and plug the power adapter into an AC outlet.
Turn on the power of the α Verb, pushing the ON/OFF button on the front panel.
Turn on the power of the amplifier/mixer, and adjust the volume.

4.2. Analog

a. Input Jack Wiring

The α Verb's [LEFT] INPUT jack is also mono input for the α Verb. If you only connect a single mono cable to the [LEFT] INPUT jack, it will be also routed automatically to the [RIGHT] INPUT. However, if we are using stereo input signals, connecting a cable to the RIGHT INPUT jack, the automatic routing will be avoided and the LEFT INPUT jack will feed only the LEFT INPUT, and the RIGHT INPUT jack will feed only the RIGHT INPUT.



b. Levels Setting

Proper setting of the input and output levels is crucial in order to achieve the maximum signal-to-noise ratio. It is possible to say that it is usually best to set both input and output level controls at 3/4 or 75% of full. This will decrease the possibility of overload distortion and keep the amount of background noise to a minimum. If the signal LED on the αVerb start lighting, signalling a process saturation, turn down the Input level or decrease the volume of the source (instrument, mixer send, etc.). If the αVerb's level is causing the mixer or amp to distort, turn the Output Level down.

c. Effects Mix Level Adjust

Whether a program contains a single effect or two or three effects, you can adjust the αVerb's [MIX] control to obtain a desirable balance between the original signal and the effect one. Turning [MIX] to the right allows you to hear more Effects; turning it to the left lets you hear more of the source signal. When hooked up to an instrument setup, such as a guitar amp, the Mix setting will typically be somewhere in the middle, balancing the effects with the sound of the source instrument. If the αVerb is connected to a mixing console's Aux Send, the [MIX] control should be set all the way to the right (effects only) so that the balance can be controlled from the board.

d. Effects Bypass

At any time you can bypass the process, there by allowing the direct signal to pass through the αVerb unchanged. This can be done in two ways:

- by turning the MIX knob all the way to the left.
- by connecting a footswitch to the [BYPASS] jack and pressing the footswitch.

On the back panel you will find a footswitch jack labelled [BYPASS]. This is a mono jack with connections for a standard footswitch. The footswitch must be plugged in before the αVerb has its power turned on: αVerb will automatically recognise the right "polarity" of the pedal.

4.3 Installation

a. Standard Use

The αVerb may be placed almost anywhere: on a table, on top of an amp, next to a mixing console. If it will be on furniture, check the rubber feet provided to the bottom of the unit. Make sure to place the αVerb's power supply away from other audio equipment that may induce fields, and away from the signal wiring. It is possible that αVerb may pick up noise fields generated by other equipment such as large power amplifiers; in this case, move the αVerb until the noise goes away.

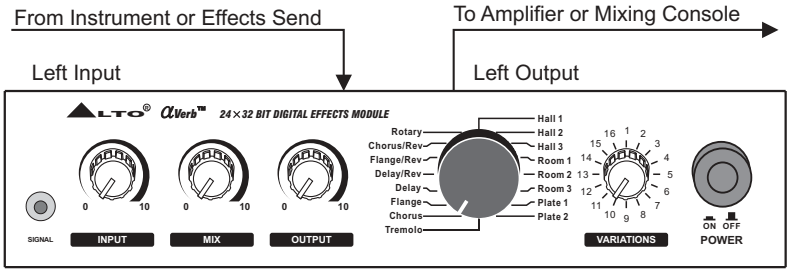
b. Application Examples

- LINE INSTRUMENT

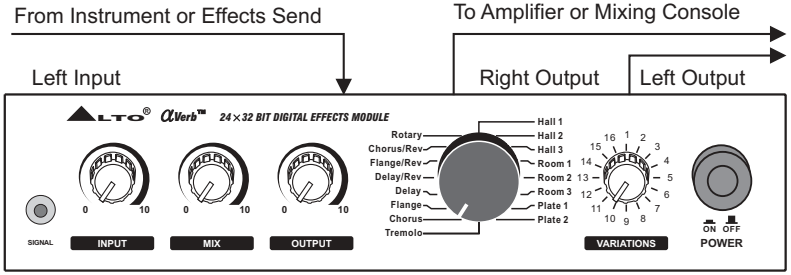
When connecting audio cables and/or turning power on and off, make sure that all devices in your system have their volume controls turned down.

The αVerb has two 1/4" unbalanced inputs and two 1/4" unbalanced outputs. These input/output configuration may provide three different audio connections options:

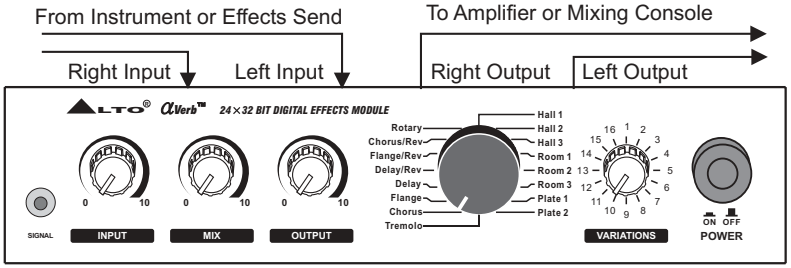
• **MONO.** Connect an audio cable to the [LEFT] INPUT of the αVerb from a mono source, and another audio cable from the [LEFT] output of the αVerb to an amplification system or mixer input.



• **MONO IN, STEREO OUT.** While still using a mono input, you could connect two audio cables to the [LEFT] and [RIGHT] outputs of the αVerb to a stereo amplification system or two mixer inputs.



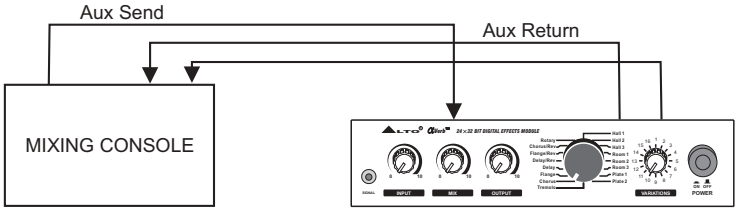
• **STEREO.** Connect two audio cables to the [LEFT] and [RIGHT] INPUTS of the αVerb from a stereo source, and two other audio cables from the [LEFT] and [RIGHT] OUTPUTS of the αVerb to a stereo amplification system or two mixer inputs.



- MIXER

Interfacing to a Mixing Console

The α Verb can accept mono or stereo sends at all system levels. The input circuitry of the α Verb can easily accept professional +8/9dBu levels while having enough input and output gain to interface with the low signal levels of home recording systems.



The α Verb may be connected to a mixing console in several different ways. It can be used to with multiple channels at once by using the auxiliary send and return controls of the mixer. Another way of interfacing is to connect the α Verb directly to the insert send and return of a single channel that is to be effected. More, α Verb could be to connected to a mixer or recording console in-line between the output of the mixing console and the input of a tape deck or power amplifier. This last setup would effect the entire mix output.

Using Aux Sends and Returns

Generally, on mixing consoles are available two types of auxiliary sends: pre-fader sends (headphone or monitor), and post-fader sends for effects units. Typically, if a mixer has more than two sends per channel (4, 6 or 8, perhaps), the first two sends are reserved for the pre-fader sends, while the remaining sends are used to send the signal to be effected to devices as the α Verb. Connect the α Verb using post-fader sends, so fading a channel out, its effects will fade also. Using a mixer's aux sends allows each channel to have its own level control going to the aux output. It is possible to mix all the channels we want to be sent to the effects by using the individual channels' aux send levels on the mixer. Most consoles also have aux master controls, which set the over-all level of each aux output. Sending signal to the α Verb is only half of the process. With a mixing console, the output of the α Verb must go back to the mixer and turned up in the mix before to be able to hear it. Depending from the mixer, there are two options for returning the effected signal to the mix:

- connecting to dedicated aux return inputs, or
- connecting to channel inputs.

Everything is easy if the mixer provides dedicated inputs (called returns) for effect devices like the α Verb. If the mixer does not have these, or the available returns have already been used all, it possible to connect the α Verb to channel inputs (if there are any remaining).

The effect returns generally should only contain effected signal, and not have any uneffected or "DRY" signal mixed with it (since these two signals are blended together at the mixer).
Therefore, it is necessary to set the mix so that only effected ("WET") signal is present at the Verb's outputs. To do this, turn the Mix control all the way to the right.

Mono in - stereo out.

If you only want to use the α Verb for a mono input signal and to connect both of its outputs back to the mixer, you will need three audio cables. Connect an audio cable from an effect send to the LEFT input of the α Verb, another 2 audio cable from the LEFT and RIGHT outputs of the α Verb to a couple of effect return or other mixer inputs. On the reverb effect α Verb creates a stereo output, even though only a single input is used.

Stereo in - stereo out.

This connection is similar to the one described above. However, by utilizing two sends from the mixer, we have to use one more audio cable to send a stereo signal to the α Verb's inputs. The use of a stereo input is especially useful on the true stereo reverb program.

How to Set Aux Send and Return Levels on the Mixer.

In the above connections, it is necessary to set proper levels on the mixer's individual Aux Sends, Aux Masters, and Aux Return masters (as well as the α Verb 's own controls) to get good, clean, quiet results.

Improper level setting is the most common cause of noise and distortion problems.

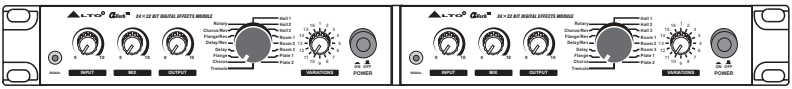
By having the correct level at every point in the send/return chain, it is possible to avoid overloading distortion and minimize noise. The most common mistake using effect units like the α Verb is to have too low the input signal level and to increase too much the output level to compensate the input and reach the desired effects level: this amplifies the noise reducing headroom. Here is a procedure that will give good results with most standard equipment:

1. Set your mixer's input levels correctly.
2. Turn up the mixer channels' AUX SEND and AUX MASTERS (if applicable) to a nominal level (this is usually between "noon" and "3:00" on a rotary knob).
3. Play the source.
4. Turn up the α Verb's [INPUT] level until you see the [SIGNAL] LED start lighting on peaks; then reduce it slightly until the led stops lighting. The ideal input level, to minimize the noise, is just below the clipping level. But if other instruments will be added to the mix later, or levels are unpredictable (as in a live show), it's preferable to leave additional headroom by turning the input level down a bit more.

5. Depending on the input sensitivity of the mixer's channels or Aux Returns, the [OUTPUT] knob of the α Verb should be set somewhere between "2:00" and fully clockwise ("5:00").
6. Turn up the AUX RETURN level until desired level of effect in the mix is reached. The control in the chain that may need to be set to a low level is the Aux Return on the mixer itself. Here is where should increase or decrease the overall effect level in the mix to minimize the noise.

4.4 Rack Mounting

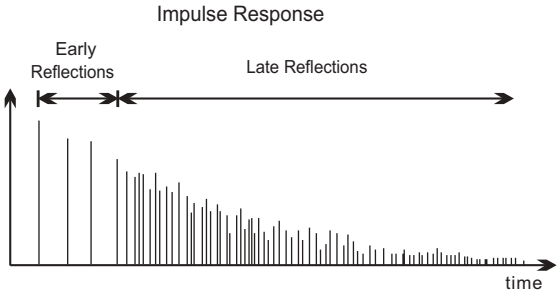
The most secure mounting is on an "universal" rack shelf, available from various rack manufacturers or music dealer. Up to two α Verb's may be mounted side-by-side in a standard universal EIA 19" equipment rack.



5. Preset Functions Descriptions

5.1 Reverbs

Reverb in nature, is the sum of a large number of distinct echoes generated by the reflection of the original sound against obstacles (i.e. walls). In a real acoustic space, the amplitude and brightness of these reflections decay over time and the decaying is depending on the room size, the position of the sound source acoustic space, the "nature" of obstacles (shape, material, dimension, etc.), and many other factors.



a. Halls

This algorithm is the simulation of a large acoustic space (as a concert hall). Halls want to simulate large rooms with many reflective surfaces, where sounds can be reflected and also hided, changing its "colour" over time. This is a classic reverb and can be used with all sound sources as vocals, drums or acoustic and electric instruments.

Hall 1 - This is a large bright hall program with 54ms predelay, and can be used for almost anything.

Hall 2 - This is a warmer hall program with 77ms predelay, and adds depth and character to acoustic instruments.
Hall 3 - The third program is a medium bright hall with no predelay, and can be used on rock snares and percussions.

b. Rooms

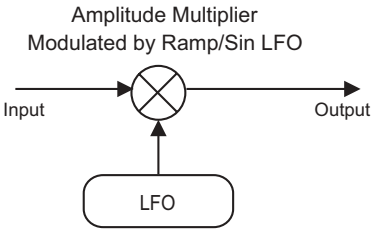
This algorithm try to reproduce the sound of a medium size room. It has a more dense and rich sound than the hall reverb algorithm, and this quality makes it good for rock and "disco" music. The attack is well defined and "aggressive", sounding very good on keyboards, guitars and drums.
Room 1- This program simulate a studio room with many "early reflection", and works well with drum sounds and acoustic instruments.
Room 2- This program is a "bright" studio room and seems to be perfect for adding a little ambience to a dry sound as the one achievable with a synth sound.
Room 3- The third room is the warmest one and is perfect for acoustic guitars and classical instruments

c. Plates

This algorithm want to simulate the "sound" of a classic plate reverb, obtained in the past using suspended sheet of metal with transducers at either end. This kind of reverb, commonly used in the 1970's, it is still useful for its transparent sound and it works well for vocals, piano, or guitar.
Plate 1- The first program is a classic bright vocal plate.
Plate 2- A warmer variation of the previous program, sounding very well on acoustic guitar and strings.

5.2 Modulations

a. Tremolo



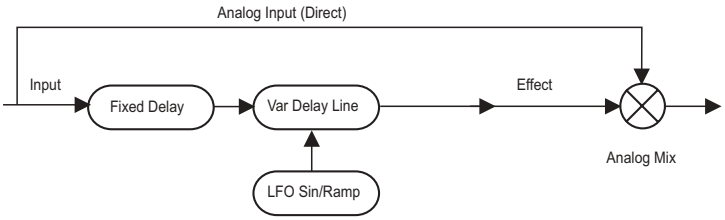
Tremolo is an amplitude modulation of the signal. It is useful for adding warmth and life to standing electric piano or guitar's chords.

Tremolo- This program provides an amplitude modulation of the input signal and is normally used as "WET" effect without adding direct sound or adding a few percentage of it, so to avoid the direct sound to cover the amplitude modulation.

Tremolo Parameters Adjust

Rate - This control sets the amplitude modulation rate.
Sin/Ramp Lfo - From 1 to 8 Ramp, From 9 to 16 Sin

b. Chorus

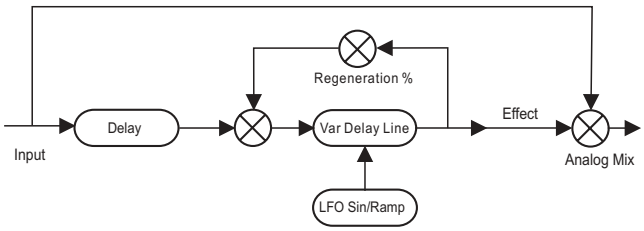


The Chorus effect tries to recreate the illusion of more than one instrument from a single instrument sound. Two musicians playing the same instrument never play in perfect unison (both time and pitch wise). In order to build up the proper illusion using an electronic device, the original sound is summed with a slightly delayed and detuned version of itself. Instead of a constant pitch deviation, more natural results come from a varying pitch deviation (two players never keep constant their relative pitch distance). α Verb's algorithm implements the variable delay and the detuning of it is modulated by an LFO (low frequency oscillator) which causes the detuning to vary. The direct sound and the detuned one are summed analogically on the outputs.

Chorus Parameters Adjust

Rate - This control sets the amplitude modulation rate.
Sin/Ramp LFO - From 1 to 8 Sin, From 9 to 16 Ramp

c. Flanger



The flanger started its life as a mechanical realization: two identical tapes were run in parallel while a human operator randomly controlled the speed of each unit, making minor variations up and down the nominal tape speed. Mixing the sound from both tapes, the signals sometimes aligned in phase, while other ti-

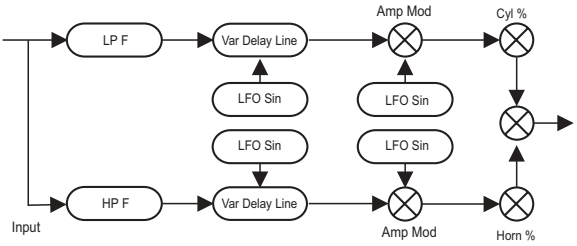
mes aligned in counter phase, resulting in a time-varying filtering that has been named 'flanger'. The structure of the flanger is then that of the mix of two randomly delayed copies of a signal. Here the detuning process is same as the one of the chorus, added with a "regeneration" part.

Flanger Parameters Adjust

Rate - This control sets the amplitude modulation rate.

Sin/Ramp LFO - From 1 to 8 Ramp, From 9 to 16 Sin

d. Rotary (Speakers)



The rotary speaker effect simulates the sound effect achieved by rotating horn speakers and a bass cylinder, as first produced for organs. The sound is altered by the Doppler effect, the directional characteristic of the speakers, phase effects due to air turbulence, etc. The rotary speaker system is normally used with organs, but can be used also for guitar amplification.

Note: When using the Rotary program, the Mix potentiometer has be turned all right on "WET" position.

Rotary Parameters Adjust

Rate - This control sets the amplitude modulation rate.

5.3 Delay

Delay effect is a single echo repetition where the repetitions occur after a certain "delay time" and where the number of repetitions depend on a "decay time", defining the time necessary to decrease the amplitude of the repetition from the original sound level to zero.

Delay- This program provides a delay of up to 1000 ms. The delay time can be adjusted in terms of delay and the decay time depends automatically from the delay time. This is a useful utility program which can add space to vocals or instruments.

Delay Parameters Adjust

Delay/Decay Time - This control sets the time between the input signal and the first delay tap and the decay time.

5.4 Combined Effects

a. Delay+Reverb - The third multieffects program adds a Room to the different delay presets.

b. Flanger+Reverb - The first multieffects program is a layered stereo flange and large room reverb. It works great on guitars, synths and electric pianos.

c. Chorus+Reverb - The second multieffects program is a layered stereo chorus and large room reverb. Also that one works great on guitars, synths and electric pianos.

5.5 Effects Summary Table

a. Program Chart

αVerb PROGRAM CHART

αVerb PROGRAM CHART		
PROGRAM NAME	DESCRIPTION	ADJUST CONTROL
HALL 1	HALL REVERB Algorithm Input LP filter: 16 kHz Process High Damp Filter: 12 kHz Process Low Damp Filter: 50 Hz Predelay Time: 0.054 ms Output LP filter: 16 kHz Output HP filter: 50 Hz	DECAY TIME
HALL 2	HALL REVERB Algorithm Input LP filter: 7 kHz Process High Damp Filter: 7.5 kHz Process Low Damp Filter: 50 Hz Predelay Time: 0.077 ms Output LP filter: 4 kHz Output HP filter: 20 Hz	DECAY TIME
HALL 3	HALL REVERB Algorithm Input LP filter: 12 kHz Process High Damp Filter: 12 kHz Process Low Damp Filter: 20 Hz Predelay Time: 0.0 ms Output LP filter: 12 kHz Output HP filter: 20 Hz	DECAY TIME
ROOM 1	ROOM REVERB Algorithm Input LP filter: 14 kHz Process High Damp Filter: 12 kHz Process Low Damp Filter: 20 Hz Predelay Time: 0.003 ms Output LP filter: 12 kHz Output HP filter: 20 Hz	DECAY TIME

ROOM 2	ROOM REVERB Algorithm Input LP filter: 16 kHz Process High Damp Filter: 14 kHz Process Low Damp Filter: 20 Hz Predelay Time: 0.003 ms Output LP filter: 12 kHz Output HP filter: 20 Hz	DECAY TIME
ROOM 3	ROOM REVERB Algorithm Input LP filter: 6.5 kHz Process High Damp Filter: 2.5 kHz Process Low Damp Filter: 20 Hz Predelay Time: 0.019 ms Output LP filter: 6 kHz Output HP filter: 200 Hz	DECAY TIME
PLATE 1	PLATE REVERB Algorithm Input LP filter: 12 kHz Process High Damp Filter: 12 kHz Process Low Damp Filter: 100 Hz Predelay Time: 0.019 ms Output LP filter: 16 kHz Output HP filter: 200 Hz	DECAY TIME
PLATE 2	PLATE REVERB Algorithm Input LP filter: 8 kHz Process High Damp Filter: 5 kHz Process Low Damp Filter: 50 Hz Predelay Time: 0.019 ms Output LP filter: 7 kHz Output HP filter: 200 Hz	DECAY TIME
TREMOLO	AMPLITUDE MODULATION Algorithm Input High Damp Filter: 18 kHz Input Low Damp Filter: 10 Hz Modulating Waves: 1 to 8 RAMP, 9 to 16 SIN	MODULATION RATE MODULATING WAVE
CHORUS	PHASE MODULATION Algorithm Input High Damp Filter: 18 kHz Input Low Damp Filter: 10 Hz Modulating Waves: 1 to 8 RAMP, 9 to 16 SIN	MODULATION RATE MODULATION DEPTH MODULATING WAVE
FLANGER	PHASE MODULATION with REGENERATION Algorithm Input High Damp Filter: 18 kHz Input Low Damp Filter: 10 Hz Modulating Waves: 1 to 8 RAMP, 9 to 16 SIN	MODULATION RATE MODULATION DEPTH REGENERATION PERCENTAGE MODULATING WAVE
DELAY	MONO DELAY Algorithm Input High Damp Filter: 18 kHz Input Low Damp Filter: 10 Hz	DECAY TIME DELAY TIME

DELAY/REVERB	MONO DELAY + REVERB Algorithm Input High Damp Filter: 18 kHz Input Low Damp Filter: 10 Hz Predelay Time: 0.0 ms Process LP filter: 12 kHz	DELAY TIME DECAY TIME
FLANGER/REVERB	MONO DELAY + FLANGE Algorithm Input High Damp Filter: 18 kHz Input Low Damp Filter: 10 Hz Predelay Time: 8 ms Process LP filter: 12kHz Modulating Waves: 1 to 8 RAMP, 9 to 16 SIN	REVERB DECAY TIME REGENERATION PERCENTAGE MODULATING WAVE
CHORUS/REVERB	MONO DELAY + CHORUS Algorithm Input High Damp Filter: 18 kHz Input Low Damp Filter: 10 Hz Predelay Time: 8 ms Process LP filter: 12kHz Modulating Waves: 1 to 8 RAMP, 9 to 16 SIN	REVERB DECAY TIME MODULATING WAVE
ROTARY SPEAKERS	STEREO LESLIE Algorithm Cylinders LP filter: 200 Hz Horns HP filter: 2 kHz Modulating Waves: SIN	ROTATION SPEED MODULATION DEPTH

b. Preset Value

Preset encoder position	Variation encoder position	Value
Hall 1	1	Decay time = 0.37 sec.
	2	Decay time = 0.46 sec.
	3	Decay time = 0.56 sec.
	4	Decay time = 0.58 sec.
	5	Decay time = 0.64 sec.
	6	Decay time = 0.75 sec.
	7	Decay time = 0.87 sec.
	8	Decay time = 0.96 sec.
	9	Decay time = 1.21 sec.

Preset encoder position	Variation encoder position	Value
	10	Decay time = 1.46 sec.
	11	Decay time = 1.90 sec.
	12	Decay time = 2.20 sec.
	13	Decay time = 3.50 sec.
	14	Decay time = 4.40 sec.
	15	Decay time = 9.10 sec.
	16	Decay time = > 30 sec.
Hall 2	1	Decay time = 0.39 sec.
	2	Decay time = 0.49 sec.
	3	Decay time = 0.51 sec.
	4	Decay time = 0.58 sec.
	5	Decay time = 0.68 sec.
	6	Decay time = 0.80 sec.
	7	Decay time = 0.82 sec.
	8	Decay time = 1.00 sec.
	9	Decay time = 1.22 sec.
	10	Decay time = 1.52 sec.
	11	Decay time = 2.11 sec.
	12	Decay time = 2.40 sec.
	13	Decay time = 3.50 sec.
	14	Decay time = 4.30 sec.
	15	Decay time = 8.60 sec.
	16	Decay time = > 30 sec.
Hall 3	1	Decay time = 0.45 sec.
	2	Decay time = 0.61 sec.
	3	Decay time = 0.73 sec.
	4	Decay time = 0.82 sec.
	5	Decay time = 1.02 sec.
	6	Decay time = 1.11 sec.
	7	Decay time = 1.19 sec.
	8	Decay time = 1.50 sec.
	9	Decay time = 1.85 sec.
	10	Decay time = 2.28 sec.
	11	Decay time = 2.80 sec.
	12	Decay time = 3.50 sec.
	13	Decay time = 5.30 sec.

Preset encoder position	Variation encoder position	Value
	14	Decay time = 6.40 sec.
	15	Decay time = 13.90 sec.
	16	Decay time = > 40 sec.
Room 1	1	Decay time = 0.32 sec.
	2	Decay time = 0.37 sec.
	3	Decay time = 0.43 sec.
	4	Decay time = 0.46 sec.
	5	Decay time = 0.55 sec.
	6	Decay time = 0.64 sec.
	7	Decay time = 0.68 sec.
	8	Decay time = 0.79 sec.
	9	Decay time = 0.99 sec.
	10	Decay time = 1.28 sec.
	11	Decay time = 1.68 sec.
	12	Decay time = 1.90 sec.
	13	Decay time = 2.90 sec.
	14	Decay time = 3.60 sec.
	15	Decay time = 6.90 sec.
	16	Decay time = > 25 sec.
Room 2	1	Decay time = 0.36 sec.
	2	Decay time = 0.43 sec.
	3	Decay time = 0.44 sec.
	4	Decay time = 0.50 sec.
	5	Decay time = 0.54 sec.
	6	Decay time = 0.59 sec.
	7	Decay time = 0.75 sec.
	8	Decay time = 0.89 sec.
	9	Decay time = 1.07 sec.
	10	Decay time = 1.33 sec.
	11	Decay time = 1.84 sec.
	12	Decay time = 2.00 sec.
	13	Decay time = 3.00 sec.
	14	Decay time = 3.60 sec.
	15	Decay time = 7.50 sec.

Preset encoder position	Variation encoder position	Value
	16	Decay time = > 25 sec.
Room 3	1	Decay time = 0.35 sec.
	2	Decay time = 0.38 sec.
	3	Decay time = 0.41 sec.
	4	Decay time = 0.47 sec.
	5	Decay time = 0.52 sec.
	6	Decay time = 0.62 sec.
	7	Decay time = 0.73 sec.
	8	Decay time = 0.87 sec.
	9	Decay time = 1.07 sec.
	10	Decay time = 1.30 sec.
	11	Decay time = 1.65 sec.
	12	Decay time = 1.80 sec.
	13	Decay time = 2.80 sec.
	14	Decay time = 3.40 sec.
	15	Decay time = 6.30 sec.
	16	Decay time = > 25 sec.
Plate 1	1	Decay time = 0.42 sec.
	2	Decay time = 0.49 sec.
	3	Decay time = 0.63 sec.
	4	Decay time = 0.64 sec.
	5	Decay time = 0.73 sec.
	6	Decay time = 0.85 sec.
	7	Decay time = 0.96 sec.
	8	Decay time = 1.09 sec.
	9	Decay time = 1.45 sec.
	10	Decay time = 1.65 sec.
	11	Decay time = 2.18 sec.
	12	Decay time = 2.50 sec.
	13	Decay time = 3.90 sec.
	14	Decay time = 4.90 sec.
	15	Decay time = 9.10 sec.
	16	Decay time = > 30 sec.
Plate 2	1	Decay time = 0.44 sec.

Preset encoder position	Variation encoder position	Value
	2	Decay time = 0.51 sec.
	3	Decay time = 0.56 sec.
	4	Decay time = 0.64 sec.
	5	Decay time = 0.67 sec.
	6	Decay time = 0.81 sec.
	7	Decay time = 0.96 sec.
	8	Decay time = 1.07 sec.
	9	Decay time = 1.39 sec.
	10	Decay time = 1.61 sec.
	11	Decay time = 2.13 sec.
	12	Decay time = 2.40 sec.
	13	Decay time = 3.70 sec.
	14	Decay time = 4.80 sec.
	15	Decay time = 8.50 sec.
	16	Decay time = > 30 sec.

Preset encoder position	Variation encoder position	Delay Time	Decay Time
Delay	1	30 ms	-
	2	40 ms	-
	3	50 ms	-
	4	60 ms	-
	5	70 ms	-
	6	80 ms	-
	7	100 ms	1.00 sec.
	8	200 ms	1.90 sec.
	9	300 ms	3.30 sec.
	10	400 ms	4.00 sec.
	11	500 ms	4.30 sec.
	12	600 ms	5.30 sec.
	13	700 ms	6.30 sec.
	14	800 ms	7.00 sec.
	15	900 ms	8.00 sec.
	16	1000 ms	10.00 sec.

Preset encoder position	Variation encoder position	Delay Time	Delay Decay Time	Rev. Decay Time
Delay+Rev	1	30 ms	-	1.30 sec.
	2	40 ms	-	1.30 sec.
	3	50 ms	-	1.30 sec.
	4	60 ms	-	1.40 sec.
	5	80 ms	0.75 sec.	1.30 sec.
	6	100 ms	1.40 sec.	2.30 sec.
	7	200 ms	2.40 sec.	3.00 sec.
	8	300 ms	3.40 sec.	3.30 sec.
	9	400 ms	3.30 sec.	4.00 sec.
	10	500 ms	4.00 sec.	4.40 sec.
	11	600 ms	4.40 sec.	6.00 sec.
	12	700 ms	5.00 sec.	7.30 sec.
	13	800 ms	6.00 sec.	7.40 sec.
	14	900 ms	7.20 sec.	8.00 sec.
	15	1000 ms	8.20 sec.	8.30 sec.
	16	1200 ms	9.00 sec.	9.30 sec.

Preset encoder position	Variation encoder position	Period	Rate	Amplitude Att.
Tremolo	1	840 ms	1.19 Hz	84 %
	2	600 ms	1.60 Hz	84 %
	3	380 ms	2.63 Hz	84 %
	4	260 ms	3.84 Hz	84 %
	5	200 ms	5.00 Hz	84 %
	6	130 ms	7.69 Hz	84 %
	7	100 ms	10.00 Hz	84 %
	8	60 ms	16.00 Hz	84 %
	9	1400 ms	0.71 Hz	36 %
	10	840 ms	1.19 Hz	36 %
	11	650 ms	1.56 Hz	36 %
	12	470 ms	2.12 Hz	36 %

Preset encoder position	Variation encoder position	Period	Rate	Amplitude Att.
	13	400 ms	2.50 Hz	36 %
	14	320 ms	3.12 Hz	36 %
	15	260 ms	3.84 Hz	36 %
	16	200 ms	5.00 Hz	36 %

Preset encoder position	Variation encoder position	Period	Rate
Chorus	1	11.00 sec.	0.09 Hz
	2	4.80 sec.	0.21 Hz
	3	3.20 sec.	0.31 Hz
	4	1.78 sec.	0.56 Hz
	5	1.20 sec.	0.83 Hz
	6	1.09 sec.	0.91 Hz
	7	0.75 sec.	1.30 Hz
	8	0.41 sec.	2.43 Hz
	9	12.80 sec.	0.07 Hz
	10	9.20 sec.	0.11 Hz
	11	1.06 sec.	0.94 Hz
	12	0.62 sec.	1.61 Hz
	13	0.47 sec.	2.13 Hz
	14	0.56 sec.	1.78 Hz
	15	0.40 sec.	2.50 Hz
	16	0.30 sec.	3.33 Hz

Preset encoder position	Variation encoder position	Period	Rate	Feedback %
Flanger	1	13.4 sec.	0.075 Hz	67
	2	13.4 sec.	0.075 Hz	75
	3	13.4 sec.	0.075 Hz	82
	4	6.05 sec.	0.165 Hz	60
	5	6.05 sec.	0.165 Hz	72
	6	6.05 sec.	0.165 Hz	82

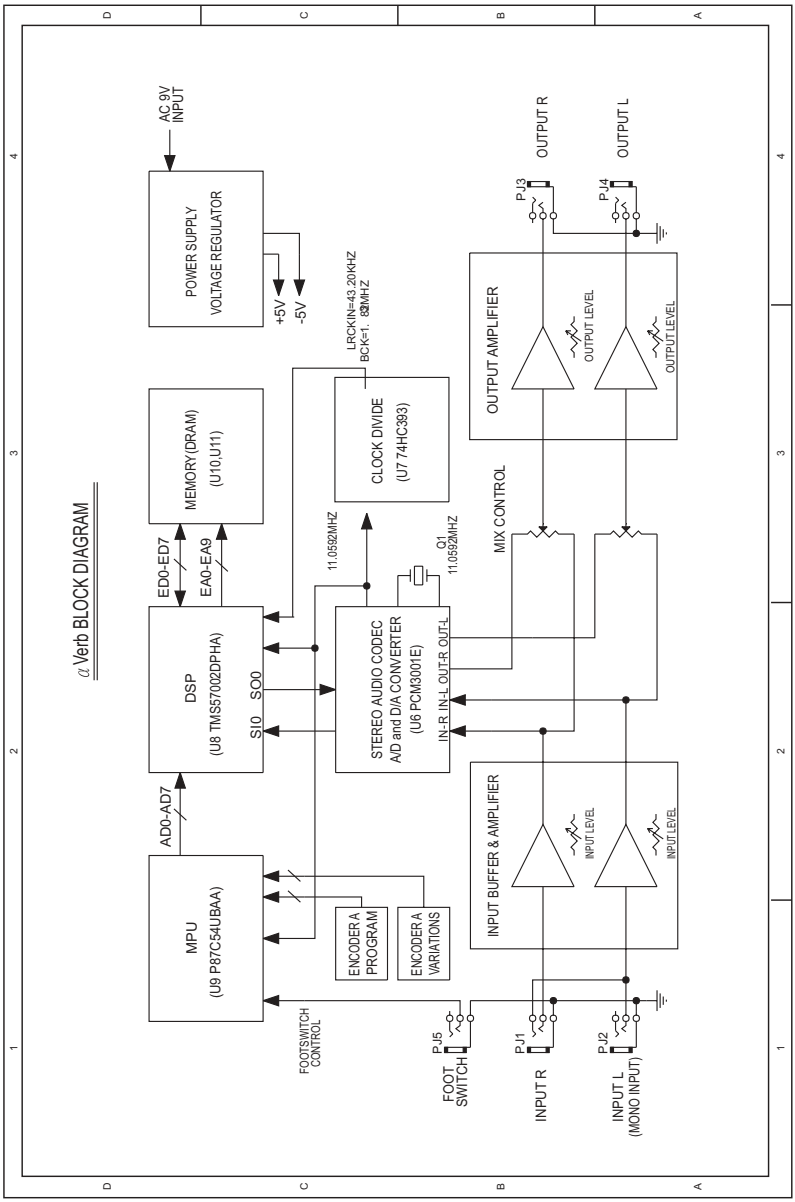
Preset encoder position	Variation encoder position	Period	Rate	Feedback %
	7	6.05 sec.	0.165 Hz	89
	8	6.05 sec.	0.165 Hz	92
	9	4.50 sec.	0.222 Hz	82
	10	3.00 sec.	0.333 Hz	82
	11	1.75 sec.	0.571 Hz	82
	12	1.30 sec.	0.769 Hz	82
	13	1.00 sec.	1.000 Hz	82
	14	0.75 sec.	1.333 Hz	82
	15	0.70 sec.	1.428 Hz	82
	16	0.53 sec.	1.886 Hz	82

Preset encoder position	Variation encoder position	Chorus Period	Chorus Rate	Rev. Decay Time
Chorus + Rev.	1	1.30 sec.	0.769 Hz	0.78 sec.
	2	1.30 sec.	0.769 Hz	1.30 sec.
	3	1.30 sec.	0.769 Hz	1.75 sec.
	4	1.30 sec.	0.769 Hz	2.20 sec.
	5	1.30 sec.	0.769 Hz	3.00 sec.
	6	1.30 sec.	0.769 Hz	4.00 sec.
	7	1.30 sec.	0.769 Hz	5.20 sec.
	8	1.30 sec.	0.769 Hz	8.00 sec.
	9	1.60 sec.	0.625 Hz	0.78 sec.
	10	1.60 sec.	0.625 Hz	1.30 sec.
	11	1.60 sec.	0.625 Hz	1.75 sec.
	12	1.60 sec.	0.625 Hz	2.20 sec.
	13	1.60 sec.	0.625 Hz	3.00 sec.
	14	1.60 sec.	0.625 Hz	4.00 sec.
	15	1.60 sec.	0.625 Hz	5.20 sec.
	16	1.60 sec.	0.625 Hz	8.00 sec.

Preset encoder position	Variation encoder position	Flanger Period	Flanger Rate	Rev. Decay Time
Flanger + Rev.	1	2.00 sec.	0.500 Hz	0.80 sec.
	2	2.00 sec.	0.500 Hz	1.30 sec.
	3	2.00 sec.	0.500 Hz	1.75 sec.
	4	2.00 sec.	0.500 Hz	2.20 sec.
	5	2.00 sec.	0.500 Hz	3.00 sec.
	6	2.00 sec.	0.500 Hz	3.60 sec.
	7	2.00 sec.	0.500 Hz	4.20 sec.
	8	2.00 sec.	0.500 Hz	6.40 sec.
	9	2.60 sec.	0.384 Hz	0.80 sec.
	10	2.60 sec.	0.384 Hz	1.30 sec.
	11	2.60 sec.	0.384 Hz	1.75 sec.
	12	2.60 sec.	0.384 Hz	2.20 sec.
	13	2.60 sec.	0.384 Hz	3.00 sec.
	14	2.60 sec.	0.384 Hz	3.60 sec.
	15	2.60 sec.	0.384 Hz	4.20 sec.
	16	2.60 sec.	0.384 Hz	6.40 sec.

6. Technical Specifications

6.1 Block Diagram



6.2 Specifications

Electrical	
Frequency Response:	+0.5 / -1.5 dB from 20Hz to 20 kHz
S/N Ratio (process)	80 dB "A" wtg, 20 Hz-22kHz
S/N Ratio (bypass)	>90 dB "A" wtg, 20 Hz-22kHz
THD+Noise:	<0.008% @ 1kHz (0dBV, bypass)
Input	
Number of Channels:	2
Format:	1/4" unbalanced
Maximum Level (bypass):	+9 dBu
Impedance:	>500 Kohms
A/D - D/A Conversions	
A/D converter:	1 bit Sigma-Delta
D/A converter:	1 bit Sigma-Delta
Output	
Number of Channels:	2
Format:	1/4" unbalanced
Maximum Level (bypass):	+9 dBu
Output Impedance:	<500 ohms
Front Panel	
Controls	IN/OUT levels (ANALOG)
	PROGRAM selections (2 knobs)
Indicators	Power, Signal clip LED
Rear Panel	
Input (LEFT/MONO, RIGHT)	1/4" 2-conductor (mono)
Output (LEFT, RIGHT)	1/4" 2-conductor (mono)
BYPASS	1/4" 2-conductor (auto-sense pedal type)
	for momentary footswitches
Power	9 Volt AC Power Transformer

Processing and Memory	
Processor Speed:	12 MIPs (million instructions per second)
Internal DSP resolution:	52 bit MPY accumulator
Main Preset Programs	16
Preset Total Combinations	256
Internal digital audio memory:	3000 milliseconds
Physical	
Net Weight:	1kg(2.20lb)
Dimension:	200(W)×150(D)×45(H)mm (7.87" × 5.91"× 1.77")

7. Warranty

1. WARRANTY REGISTRATION CARD

To obtain Warranty Service, the buyer should first fill out and return the enclosed Warranty Registration Card within 10 days of the Purchase Date.

All the information presented in this Warranty Registration Card gives the manufacturer a better understanding of the sales status, so as to purport a more effective and efficient after-sales warranty service.

Please fill out all the information carefully and genuinely, miswriting or absence of this card will void any of your warranty service.

2. RETURN NOTICE

2.1 In case of return for any warranty service, please make sure that the product is well packed in its original shipping carton, and it can protect your unit from any other extra damage.

2.2 Please provide a copy of your sales receipt or other proof of purchase with the returned machine, and give detail information about your return address and contact telephone number.

2.3 A brief description of the defect will be appreciated.

2.4 Please prepay all the costs involved in the return shipping, handling and insurance.

3. TERMS AND CONDITIONS

3.1 ▲LTO warrants that this product will be free from any defects in materials and/or workmanship for a period of 1 year from the purchase date if you have completed the Warranty Registration Card in time.

3.2 The warranty service is only available to the original consumer, who purchased this product directly from the retail dealer, and it can not be transferred.

3.3 During the warranty service, ▲LTO may repair or replace this product at its own option at no charge to you for parts or for labor in accordance with the right side of this limited warranty.

3.4 This warranty does not apply to the damages to this product that occurred as the following conditions:

- Instead of operating in accordance with the user's manual thoroughly, any abuse or misuse of this product.
- Normal tear and wear
- The product has been altered or modified in any way .
- Damage which may have been caused either directly or indirectly by another product/ force/etc.
- Abnormal service or repairing by anyone other than the qualified personnel or technician.

And in such cases, all the expenses will be charged to the buyer.

3.5 In no event shall ▲LTO be liable for any incidental or consequential damages. Some states do not allow the exclusion or limitation of incidental or consequential damages, so the above exclusion or limitation may not apply to you.

3.6 This warranty gives you the specific rights, and these rights are compatible with the state laws, you may also have other statutory rights that may vary from state to state.

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